AMENDMENTS TO THE CLAIMS

1. (Currently amended) A method of auto-calibrating a surround sound system, comprising the acts of:

producing an electric calibration signal, said calibration signal being a temporal maximum length sequence (MLS) signal,

supplying said calibration signal to an electro-acoustic converter for converting the calibration signal to an acoustic response,

transmitting the acoustic response as a sound wave in a listening environment to an acousto-electric converter for converting the acoustic response received by the acousto-electric converter to an electric response signal,

~~detecting in~~ correlating the electric response signal with the MLS signal to determine an impulse response,

~~a reflected signal and isolating a portion of the response signal~~ determining from the impulse response an anechoic portion of the impulse response between a time of flight signal and [[the]] a first reflected signal,

~~correlating~~ using the ~~isolated~~ anechoic portion of the impulse ~~electric~~ response ~~signal with the electric calibration signal~~ to compute filter coefficients, and

processing the filter coefficients together with a predetermined channel response of the electro-acoustic converter to produce a substantially whitened system response.

2. (Original) The method of claim 1, wherein the acoustic response is radiated in the listening environment for a time less than approximately 3 seconds.

3. (Original) The method of claim 1, wherein the surround sound system includes a plurality of audio channels, with each channel having at least one electro-acoustic converter, wherein the substantially whitened response is produced independently for each audio channel.

4. (Currently amended) A method of optimizing a matched filter for whitening an audio channel in a listening environment, comprising:

a. producing in the audio channel a test output sound corresponding to a temporal maximum length sequence (MLS) signal,

b. receiving the test output sound at a predetermined location in the listening environment~~, thereby producing~~ and correlating the received signal with the MLS signal to produce an impulse response,

~~c. analyzing a correlation between the impulse response and the MLS signal,~~

[[d.]] c. generating filter coefficients of the matched filter,

[[e.]] d. repeating steps (a) through (c) [[(d)]] with at least one other MLS signal having a different temporal maximum length, and

[[f.]] e. optimizing the matched filter by selecting those generated filter coefficients that minimize an error term between a desired filter response of the matched filter producing the whitened audio channel and the filter response produced with the generated filter coefficients when driven by the corresponding maximum length MLS signal.

5. (Previously presented) The method of claim 4, wherein the filter coefficients represent coefficients of a polynomial model of the impulse response.

6. (Previously presented) The method of claim 4, wherein the filter coefficients are generated by an auto regressive (AR) model.

7. (Original) The method of claim 5, wherein generating the filter coefficients includes optimizing a closeness of fit between the polynomial model and the matched filter.

8. (Canceled)

9. (Original) The method of claim 5, further comprising cascading the matched filter with a useful audio signal so as to produce the substantially whitened audio channel.

10. (Previously presented) An auto-calibrating surround sound (ACSS) system, comprising:

an electro-acoustic converter disposed in an audio channel and adapted to emit a sound signal in response to an electric input signal,

a processor generating a test signal represented by a temporal maximum length sequence (MLS) and at least one other test signal represented by a different temporal maximum length sequence, and the processor supplying the test signals as electric input signals to the electro-acoustic converter,

an acousto-electric converter receiving the sound signal in a listening environment and supplying received electric signals to the processor, and

a coefficient extractor which generates filter coefficients of a corrective filter,

wherein the processor correlates the received electric signals with the test signals and optimizes the corrective filter by selecting those generated filter coefficients that minimize an error term between a desired filter response of the corrective filter that produces a whitened audio response of the audio channel in the listening environment and the filter response produced with the generated filter coefficients, when driven by the corresponding MLS signal.

11. (Previously presented) The ACSS system of claim 10, wherein the processor includes an impulse modeler that produces a polynomial least-mean-square (LMS) error fit between a desired whitened response and the filter response produced with the generated filter coefficients.

12. (Canceled)

13. (Previously presented) The ACSS system of claim 10, wherein the corrective filter is located in an audio signal path between an audio signal line input and the electro-acoustic converter and cascaded with the audio signal line input.

14. (Previously presented) The ACSS system of claim 10, wherein the corrective filter forms a part of the processor.

15. (Previously presented) The ACSS system of claim 10, wherein the processor is a digital signal processor (DSP).

16. (Original) The ACSS system of claim 15, further including an analog-to-digital (A/D) converter that converts an analog audio line input and the electric signal supplied by the acousto-electric converter into temporal digital signals.

17. (Original) The ACSS system of claim 15, further including a digital-to-analog (D/A) converter that converts digital output signals from the DSP to an analog audio line output for driving the electro-acoustic converter.

18. (Canceled)

19. (Currently amended) An auto-calibrating surround sound (ACSS) system, comprising:

an electro-acoustic converter disposed in an audio channel and adapted to emit a sound signal in response to an electric input signal,

a processor generating a test signal represented by a temporal maximum length sequence (MLS) and supplying the test signal as the electric input signal to the electro-acoustic converter, and

an acousto-electric converter receiving the sound signal in a listening environment and supplying a received electric signal to the processor,

wherein the processor correlates ~~detects in~~ the received electric signal with the MLS sequence to compute an impulse response, determines from the impulse response ~~a reflected signal and correlates a portion of the response signal between~~ a time of flight signal and [[the]] a first reflected signal, thereby defining an anechoic portion of the impulse response, computes ~~with the test signal to compute~~ filter coefficients from the anechoic portion of the impulse response, ~~said processor processing~~ and processes the filter coefficients together with a predetermined channel response of the electro-acoustic converter to produce a substantially whitened system response.

20. (Previously presented) The ACSS system of claim 19, wherein the processor includes an impulse modeler that produces a polynomial least-mean-square (LMS) error fit between a desired whitened response and the substantially whitened response determined from the correlated signals.

21. (Previously presented) The ACSS system of claim 19, further comprising a coefficient extractor which generates filter coefficients of a corrective filter to produce the substantially whitened response of the audio channel.

22. (Previously presented) The ACSS system of claim 21, wherein the corrective filter is located in an audio signal path between an audio signal line input and the electro-acoustic converter and cascaded with the audio signal line input.

23. (Previously presented) The ACSS system of claim 21, wherein the corrective filter forms a part of the processor.

24. (Previously presented) The ACSS system of claim 19, wherein the processor is a digital signal processor (DSP).

25. (Previously presented) he ACSS system of claim 24, further including an analog-to-digital (A/D) converter that converts an analog audio line input and the electric signal supplied by the acousto-electric converter into temporal digital signals.

26. (Previously presented) The ACSS system of claim 24, further including a digital-to-analog (D/A) converter that converts digital output signals from the DSP to an analog audio line output for driving the electro-acoustic converter.

Add new claims 27-28:

27. (New) The method of claim 4, further comprising before step (c): analyzing the impulse response and determining an anechoic portion of the impulse response located between a time of flight signal and a first reflected signal, and generating the filter coefficients of the matched filter from the anechoic portion.

28. (New) The system of claim 10, wherein the coefficient extractor generates filter coefficients from an anechoic portion of the impulse response located between a time of flight signal and a first reflected signal, and generates the filter coefficients of the corrective filter from the anechoic portion.